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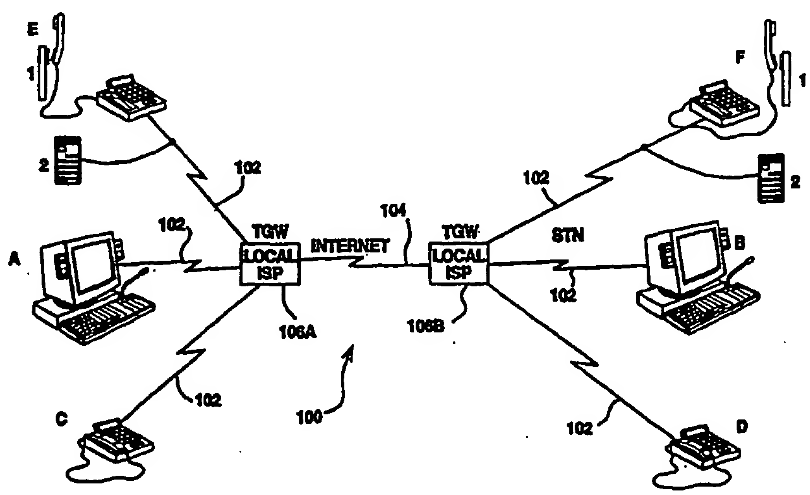

(57) Abstract

In one aspect, the present invention is directed to a method for audio communication over an integrated STN-computer. The method includes the steps of identifying an originated audio communication as either an analog audio communication or an audio over data path communication and transmitting said audio communication in accordance with its identification. The invention also discloses a system same which a plurality of audio communication devices, local and remote Internet Service Providers (ISP) connected to each of said plurality of audio communication devices by an STN portion, a data packet based Internet portion connecting said local and remote Internet Service Providers (ISP), and identification means for identifying said plurality of audio communication devices, as either an analog audio communication or an audio over data path communication, to said local and remote ISP. In yet another aspect, the invention provides a method, which may or may not be employed with the method and system disclosed above, for saving INTERNET Access Time which includes the step of performing at least part of the modem start up stages with at least part of the actual access stage.

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SYSTEM AND METHOD FOR PROVIDING AUDIO COMMUNICATION OVER AN INTEGRATED SWITCHED TELEPHONE AND COMPUTER NETWORK

BACKGROUND OF THE INVENTION

Audio communication over a computer network, such as over the INTERNET is known in the art. For example, the INTERNET PHONE™ commercially available from VocalTec Ltd. of Herzelia, Israel the assignee of the present invention, enables INTERNET users audio communication in a voice over data path via the INTERNET. Such users are equipped with standard PCs with an audio card, such as a Sound Blaster commercially available from Creative Laboratories Inc. of California, USA, with microphone and speakers.

Recently, is also became possible to call an INTERNET PHONE™ user from a conventional telephone using a TELEPHONY GATEWAY also commercially available from VocalTec Ltd. of Herzelia, Israel the assignee of the present invention. In this state of the art systems schematically illustrated in Fig. 1, a conventional telephone user 10 is connected via the conventional Switched Telephone Network (STN) 12 and Telephone Central Offices (CO) 14A and 14B to other telephone users, such as user 15 and via a TELEPHONY GATEWAY 16A and the INTERNET 17 and Gateway 16B to an INTERNET user equipped with PC 18 equipped with an INTERNET PHONE™. These state of the art systems are limited in the types of audio communications they support and require additional circuits in the STN to fully support audio communications.

SUMMARY OF THE INVENTION

The present invention provides a system and method which provides audio communication over an integrated switched telephone network (STN) and computer network.

According to one aspect of the invention, the network distinguishes analog audio communications from audio over a data path communications so as to enable outgoing and incoming communications between any audio supporting devices (conventional telephone, PC). Each audio communication initiator identifies the call as an analog audio communication (also termed circuit mode communication) or an audio over a data path communication thus enables the network to better control call routing in the integrated STN-computer network.

An advantage of the present invention is that it enables to support both analog audio and audio over a data path audio communications using the same circuit on the STN.

According to one embodiment, the identification and handshake between the audio communication initiator and the network is done using the V.8 or the V.8bis protocols specifically using a portion thereof termed herein etiquettes.

In one embodiment, the present invention provides a method for audio communication over an integrated STN-computer network which includes the steps of identifying an originated audio communication as either an analog audio communication or an audio over data path communication and transmitting said audio communication in accordance with its identification.

In a preferred embodiment, the identifying includes employing a communication protocol utilized to establish the audio communication to provide said identification. Preferably, but not necessarily, the protocol is the V.8 or V.8bis protocol.

In another embodiment, the present invention provides a system for audio communication over an integrated STN-computer network. The system preferably includes a plurality of audio communication devices, local and remote

Internet Service Providers (ISP) connected to each of said plurality of audio communication devices by an STN portion, a data packet based Internet portion connecting said local and remote Internet Service Providers (ISP), and identification means, for identifying said plurality of audio communication devices, as either an analog audio communication or an audio over data path communication, to said local and remote ISP.

In a preferred embodiment, the identification means include a communication protocol, preferably but not necessarily the V.8 or V.8bis protocol. In one embodiment, for example when the call is initiated from a computer, the V.8 or V.8bis signals are generated by the audio communication device itself. Alternatively, the V.8 or V.8bis signals are generated by a dedicated device. In yet another aspect of the invention, there is provided a method for saving INTERNET Access Time (IAT) which includes the step of performing at least part of the modem start up stages with at least part of the actual access stage.

In an embodiment of the invention, the ASCII log stage of said actual access steps is performed at least partially concurrently with the network interaction step of said modem start-up stage.

BRIEF DESCRIPTION OF THE DRAWINGS AND APPENDIX

The present invention will be understood and appreciated more fully from the following detailed description taken in conjunction with the drawings in which:

Fig. 1 is a schematic illustration of a prior art network;

Fig. 2 illustrates an integrated STN-computer network constructed and operative in accordance with the present invention; and

Figs. 3A - 3C are schematic timing diagram on the same schematic horizontal time scale illustrating the sequence of prior art IAT stages (Fig. 3A) and the at least partial overlap of certain IAT stages in accordance with the present invention (Figs. 3B and 3C).

Appendix A: Draft Recommendation V.8 "Procedures for Starting Sessions of Data Transmission Over the General Switched Telephone Network";

Appendix B: Implementation details of non-standard information fields in V.8.

DETAILED DESCRIPTION OF THE PRESENT INVENTION

Fig. 2 illustrates an integrated STN - Computer network, generally referenced 100 constructed and operative in accordance with the present invention. Network 100 includes any number of audio communication devices capable of communicating with any other device over the network by identifying themselves as an analog audio communication or as an audio over a data path communication.

Network 100 comprises an STN portion 102 and a data packet based (INTERNET) portion 104 connected therebetween by INTERNET Service Providers (ISP) servers 106 and 106 B equipped with a TELEPHONY GATEWAY bridging the STN and the packet portions of the network. A preferred TELEPHONY GATEWAY is the one commercially available from VocalTec Ltd. of Herzelia, Israel.

In the present invention, users, such as users A to F in the illustrated embodiment conduct audio communications therebetween via the STN and INTERNET by identifying themselves to the ISP 106 A and 106B as analog audio or audio over data path communication. This enables ISPs 106 to provide full audio services, both for the communication originators and the audio communication recipients.

In each communication, the communication originator identifies himself as an analog audio communication or as audio over data path communication which may be in circuit or packet mode. ISP 106A handles the communication accordingly as detailed herein below. If the receiving end ISP 106B (assuming a call initiated from A, C or E to B, D or F) uses generally similar indications to identify the type of call to the call recipient. This allows the call recipient to select which mode of operation is most desirable.

In one embodiment of the present invention the etiquettes portion of the V.8 or V.8bis protocols are used to provide the audio communication identification handshake as described herein below. In other embodiments, the invention is

used to provide additional information, such as billing information, as detailed herein below.

The following examples illustrate different modes of communication allowed by the present invention and illustrated in Fig. 2.

A to B

A initiates an audio "call" to B by establishing/using STN connection to local ISP 106A;

A passes B's address information as data to local ISP 106A;

Local ISP 106A routes "call" to remote ISP 106B;

Remote ISP 106B dials using modem over STN to B;

Modem at remote ISP 106B uses V.8 or V.8ibs etiquette to establish connection to B

Information in etiquette identifies call as audio over data path;

Upon identifying the call as audio, B brings up the software required to support audio operation;

An audio connection (bits from A to B) between A and B is established.

C to B

C (telephone capable of audio only) dials local ISP 106A;

C uses DTMF signaling or operator assistance to pass address information and identify class of call to local ISP;

Local ISP 106A acknowledges proper receipt of type of call and signals C to the same effect;

Local ISP 106A routes "call" to remote ISP 106B;

Remote ISP 106B dials using modem over STN to B;

Modem at remote ISP 106B uses V.8 or V.8bis etiquette to establish connection;

Information in etiquette identifies call as audio;

Upon identifying the call as audio, B brings up the software (different from A to B example) required to support audio operation;

An audio connection (analog voice from A to local ISP and bits from local ISP to B) between A and B is established.

A to D/F

A initiates "call" to D/F by establishing/using STN connection to local ISP 106A.

A passes D/F address information as data over modem to local ISP 106A and connection is audio over data path.

Local ISP 106A routes "call" to remote ISP 106B. There two possibilities based on above:

Remote ISP converts Internet bit stream to audio and dials using DTMF or pulse signaling D/F address.

D/F answers call "Hello" and end-to-end voice path is established, analog voice over each access section and bits over the Internet.

E to B

User initiates call using etiquette signaling box (E1 or E2) identifying voice only capability and providing B address information to local ISP (new). In the E1 case, the signaling box is an acoustic device 112 which generates V.8 signals and sends them to the hand set microphone of telephone 113. In the E2 case, the signaling box is a device 114 which sends the V.8 signals directly over the analog line 102;

Local ISP 106A acknowledges proper receipt of class of call and signals E to same effect;

Local ISP 106A routes "call" to remote ISP 106B;

An audio connection (analog voice from E to local ISP 106A and bits from remote ISP 106B to B) between E and B is established.

In each of the above examples the addressing information may also contain:

1. Class of call (e.g., circuit mode or packet mode);
2. QoS (related to but separate from 1) includes delay, codec choice, etc.;
3. Billing information (e.g., smart card, telephone credit card);
4. Addressing (including multiple addresses for conferencing);

5. Control signals from end-to-end.

V.8 or V.8bis (Start-up [etiquette] protocols defined by the ITU-T Study Group 14) (Appendix A) are the preferred, however non limiting, means to identify that a specific circuit mode call is in fact a call that is extended to/from a packet mode network and transfer the required/desired information between the ISP and the user as well as end-to-end. The invention described requires a new code point in V.8bis to indicate voice Internet access or a new mechanism in V.8. Initially the V.8bis code point could be passed as a unique code by specifying a Non Standard Network (V.8bis Table 5.3, Network Type Coding) or as a Non Standard Field (Table 5.1, Identification Field) depending on compatibility with the existing V.8bis implementations. At a later stage, this can be converted to an ITU standard.

The following comments are made with respect to the examples provided above.

Example 1

Internet Call A to B

This is the mode most commonly used today when the Internet call is extended over the STN. The data path through the Internet is extended over the STN using IETF defined protocols such as PPP (point-to-point protocol). PPP is passed over the STN using telephone data modems (e.g., V.32, V.34). These data calls use a browser (name of a software program for accessing the Internet), such as the one commercially available from Microsoft of Washington, USA, and can provide voice capability via an additional software program (e.g., Internet Phone commercially available from Vocaltec Ltd. of Herzelia, Israel).

At the answering end, the use of V.8 or V.8bis allows the answering modem to request the attached host to bring up the voice over Internet application. This is necessary as the answering end has no other way to identify that the call is a voice over Internet call prior to the data path being put through (which requires that the host based voice software application and related IP stacks be running).

Example 2

Voice call initiated via A and answered via the telephone (D)

The answering user hears the telephone ring and manually picks up the receiver. The answering end user hears the calling tone (V.8) or ES MS (V.8bis). There are two choices:

1) The answering party does not want to use a computer. In this case they signal with hook flash or DTMF # (which is received at the answering end ISP modem and converted to a message that is sent over the data channel back to the originating end user modem/host) that they only support audio communications.
2) The answering end user wished to have more than audio capabilities and directs their computer (assumes user at D has both telephone and computer connected to the STN) to answer the call.

Other variations

If A knows ad priori that location D is a telephone, which only supports analog voice, then A could, if A shows the ability to transmit/receive audio, use etiquette negotiation to request analog voice to local ISP. This eliminates modem start-up delay of 8-20 seconds and likely provides better voice quality connection (less delay and one digital voice path, non-tandem). It also means that the "call" can be extended from the remote ISP to D without any computer tones heard by the user at D (just like a telephone call). This counter intuitive example demonstrates the value in identifying analog voice as a separate service to audio over a data stream.

Example 3

Voice call Origination from a telephone [C]. Telephone answering is discussed under Example 2.

This call cannot be implemented using existing ISP data modems connected to incoming lines unless there is a means to notify the access ISP

router that the access call is analog voice. There are three possible ways to accomplish this without ISP circuits or equipment:

a. Indicate request for operator via sending "0".
b. Send DTMF signaling of call type and addressing.
c. Use attachment to the telephone (acoustic or direct connect (E1 & E2)) to send V.8 or V.8bis signals, which would identify call type and addressing.

In case c, a specific code in V.8 or V.8bis is preferably passed via the calling tone or via ES MS (in V.8bis) which would allow the ISP to connect the call via the Internet to the remote ISP which would then dial the remote end originating modem.

The above examples can all be implemented using the V.8 or V.8bis etiquette.

More Detailed Description of the Use of V. 8. Etiquette

ITU Recommendation V.8 etiquette with modifications can allow negotiation of analog voice access as an alternative to data modem (audio over data). An enablement of the patent using Recommendation V.8bis is also practical but not discussed. However, V.8bis, which is duplex, cannot be implemented with a transmit only (single cup) acoustic coupler etiquette signaling device.

1. A call is initiated from A to the local ISP in this example.

2a. After one second the modem/device at A transmits the Call Indicator signal (CI, defined in V.8) which includes an unused GSTN access category octet, e.g., 010110xx11 to indicate that the non-standard analog access is desired. Following this CI signal is the information field defined below. This is a non-standard implementation.

2b. As an alternative, a V.8 call function octet could be standardized as a non-standard facilities (NSF) designator. Following this CI signal is the information field defined below. This could become a formal standard implementation.

3. The local ISP modem/device answers the call, receives the CI plus information field sequence and initiates an ANSam signal (defined in V.8) which is switched on and off at a one second rate (or any rate discernible to a human) - called shutter AMF to indicate that the CI plus information field sequence was received correctly.
4. If the A end modem/device does not received the stutter ANSam signal but an ANSam signal it pauses for Te time (defined in V.8) and then sends a CM signal (defined in V.8) similar to the CI signal to negotiate a standard data modem alternative, if possible.

The information field may consist of a subset of the following:

1. An information field length indicator.
2. country code (T.35 based) (required for NSF only).
3. A provider length code (required for NSF only).
4. A Vocaltec Company code (ANSI registered or similar) (required for NSF only).
5. Class of call (e.g., circuit mode or packet mode, audio or analog access, etc.).
6. QoS (related to but separate from 1) may include delay or codec choice.
7. Billing information (e.g., smart card, telephone credit card).
8. Addressing (including multiple addresses for conferencing).
9. Control signals from end-to-end.
10. Frame check sum (two octet).

The maximum length of CI plus the information field is 2.0s to maintain compatibility with existing V.8 implementations. The CI plus information field sequence is repeated up to three times with off periods of 0.4 seconds.

Relevant portions of the ITU V8 document are appended hereto as Appendix A.

APPENDIX A

INTERNATIONAL TELECOMMUNICATION UNION

ITU-T V.8

TELECOMMUNICATION STANDARDIZATION SECTOR OF ITU

09/94)

DATA COMMUNICATION OVER THE TELEPHONE NETWORK

PROCEDURES FOR STARTING SESSIONS OF DATA TRANSMISSION OVER THE GENERAL SWITCHED TELEPHONE NETWORK

ITU-T Recommendation V.8

(Previously "CCITT Recommendation")

FOREWORD

The ITU-T (Telecommunication Standardization Sector) is a permanent organ of the International Telecommunication Union (ITU). The ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Conference (WTSC), which meets every four years, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Helsinki, March 1-12, 1993).

ITU-T Recommendation V.8 was prepared by ITU-T Study Group 14 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 20th of September 1994.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

CONTENTS

Recommendation V.8

PROCEDURES FOR STARTING SESSIONS OF DATA TRANSMISSION OVER THE GENERAL SWITCHED TELEPHONE NETWORK

(Geneva, 1994)

The ITU-T,

considering

(a) that many DCEs can provide operation according to a number of different V-Series modem Recommendations, and a means is needed to determine automatically, prior to initiation of modem handshake, the best available operational mode between two DCEs connected via the GSTN;

(b) that circuit multiplication equipments (CMEs) in the GSTN will increasingly need to deploy demodulation/remodulation in order to maintain and improve data signalling rates, and therefore need timely information on the V-Series modulation to be employed in any new session of data transmission;

(c) that there is increasing demand for the means to enable a GSTN call to be passed on automatically to an appropriate DCE;

(d) that a new start-up procedure should have the minimum potential for adversely affecting existing DCEs;

(e) that the procedure must continue to provide signals for interacting with GSTN echo-control equipment, as provided for in Recommendation V.25;

recommends

the procedures defined below.

1 Scope

This Recommendation defines signals to be exchanged between DCEs over the GSTN when a session of data transmission needs to be established, but before signals are exchanged which are specific to a particular modem Recommendation.

2 References

– ITU-T Recommendation V.2

- ITU-T Recommendation V.17
- ITU-T Recommendation V.18 (1994)
- ITU-T Recommendation V.21
- ITU-T Recommendation V.22
- ITU-T Recommendation V.22 *bis*
- ITU-T Recommendation V.23
- ITU-T Recommendation V.25
- ITU-T Recommendation V.26 *bis*
- ITU-T Recommendation V.26 *ter*
- ITU-T Recommendation V.27 *ter*
- ITU-T Recommendation V.29 (with GSTN usage defined in a T-Series Recommendation)
- ITU-T Recommendation V.32
- ITU-T Recommendation V.32 *bis*
- ITU-T Recommendation V.34 (1994)
- ITU-T Recommendation V.42
- ITU-T Recommendation T.30

3 Definitions

call indicator signal (CI): A signal transmitted from the call DCE to indicate the general communication function. CI is transmitted with an ON/OFF cadence defined in 7.1. The ON periods consist of a repetitive sequence of bits at 300 bit/s, modulating V.21(L), the low-band channel defined in Recommendation V.21.

signal CNG: The call tone defined in Recommendation T.30.

signal CT: Any call tone allowed for in Recommendation V.25.

call menu signal (CM): A signal (see 7.3) transmitted from the call DCE primarily to indicate modulation modes available in the call DCE. CM consists of a repetitive sequence of bits at 300 bit/s, modulating V.21(L), the low-band channel defined in Recommendation V.21.

CM terminator (CJ): A signal which acknowledges the detection of a JM signal and indicates the end of a CM signal. CJ consists of three consecutive octets of all ZEROs with start and stop bits, modulating V.21(L) at 300 bit/s.

joint menu signal (JM): A signal (see 7.4) transmitted from the answer DCE primarily to indicate modulation modes available jointly in the call and answer DCEs. JM consists of a repetitive sequence of bits at 300 bit/s, modulating V.21(H), the high-band channel defined in Recommendation V.21.

ANS: Answer tone as defined in Recommendation V.25.

ANSam: A sinewave signal at 2100 Hz, amplitude-modulated, as defined in 7.2.

sigC: A signal transmitted by a call DCE specific to a V-Series modem Recommendation.

sigA: A signal transmitted by an answer DCE specific to a V-Series modem Recommendation.

4 Overview of the Recommendation

4.1 Call indication

The V.8 alternative to call tone (CT), signal CI, carries information to permit the selection of call functions, eg. facsimile or data. The subsequent CM/JM exchange also provides for this function as well as other functions outlined in 4.2.

4.2 Modulation mode selection

The exchange of call menu and joint menu signals, CM and JM, enables DCEs to choose, for a forthcoming data session on the GSTN, the best V-Series modulation mode from those available in both the call and answer DCEs.

The CM/JM exchange also provides for protocol selection and GSTN access indication.

JM signals also provide information in a form suitable for GSTN circuit multiplication equipment (CME) prior to the onset of modem training.

5 Coding format

Signals CI, CM and JM use a common coding format. Each of these signals consists of a repeated sequence of bits. A sequence consists of 10 ONEs followed by 10 bits for synchronization and then information-bearing octets, each octet being preceded by a start-bit (ZERO), and followed by a stop-bit (ONE).

To avoid confusion of signal JM with Recommendation T.30 signals which also use V.21(H) modulation, a coding constraint is maintained which ensures that HDLC flags (01111110) cannot appear in the bit stream.

Each octet lies within one "information category". The coding format allows information categories to be expanded for special applications, while keeping signals brief for the majority of applications.

The first information category in a sequence shall be the call function. No particular ordering is required for subsequent information categories. All information within one category shall be carried in one octet or, when necessary, in an ordered sequence of octets.

5.1 Category octets

Category octets are those which occur first in any new information category, and include a 4-bit code identifying the information category. The format is shown below with bits listed from left to right in order of transmission:

start-bit (0) b0 b1 b2 b3 0 b5 b6 b7 stop-bit (1)

Bits b0-b3 make up the category tag with b0 the least-significant bit, b4 is set to ZERO to prevent flag simulation, and bits b5-b7 are "option bits" relevant to the information category.

5.2 Extension octets

When 3 option bits are inadequate for a particular category, any number of extension octets may follow directly after a category octet. The format for an extension octet is shown below:

start-bit (0) b0 b1 b2 0 1 0 b6 b7 stop-bit (1)

Bits b0-b2, b6 and b7 provide five additional option bits in the current category. Bit b4 is set to ONE in order to distinguish an extension octet from a category octet, and bits b3 and b5 are set to ZERO to prevent flag simulation.

6 Code tables

Information categories and extension octets beyond those specified in the tables below are all reserved for future definition by the ITU-T. To be compatible with future versions of Recommendation V.8, a receiver shall ignore all bits, codes and octets reserved for such future definition.

Table 1 shows the preamble to each signal sequence. This consists of ten ONEs followed by ten bits for synchronization.

TABLE 1/V.8

Preamble

1	1	1	1	1	1	1	1	1	1	Ten ONEs preceding each information sequence
0	0	0	0	0	0	0	0	0	1	Synchronization for CI sequences
0	0	0	0	0	0	1	1	1	1	Synchronization for CM and JM sequences

Table 2 lists the information categories, identified by a 4-bit category tag b0-b3.

TABLE 2/V.8

Information categories

Start	b0	b1	b2	b3	b4	b5	b6	b7	Stop	Category octets (b4 = 0) with category given by tag b0-b3
0	1	0	0	0	0	x	x	x	1	Call function
0	1	0	1	0	0	x	x	x	1	Modulation modes
0	0	1	0	1	0	x	x	x	1	Protocols
0	1	0	1	1	0	x	x	x	1	GSTN access

6.1 Call functions

Table 3 shows how the 3 option bits in a call-function octet are used to identify particular call functions:

TABLE 3/V.8

The call function category

Start	b0	b1	b2	b3	b4	b5	b6	b7	Stop	Octet – 'callf0'
0	1	0	0	0						Tag b0-b3 indicating the call function category
					0					Indicates a category octet
						0	0	0		To be determined by the ITU-T
						1	0	0		To be determined by the ITU-T
						0	1	0		Textphone according to Recommendation V.18
						1	1	0		To be determined by the ITU-T
						0	0	1		To be determined by the ITU-T
						1	0	1		To be determined by the ITU-T
						0	1	1		Transmit and receive data
						1	1	1		Call function as indicated in an extension octet
									1	Stop bit

6.2 Modulation modes

Table 4 shows the coding over 3 octets to indicate availability of GSTN V-Series modulation modes. Availability shall be shown only if the modulation mode can be used with the indicated call function, and if it is desired to convey that capability to the remote DCE.

TABLE 4/V.8

Modulation modes

Start	b0	b1	b2	b3	b4	b5	b6	b7	Stop	Octet – 'modn0'	Item
0	1	0	1	0						Tag b0-b3 indicating the modulation modes category	
					0					Indicates a category octet	
						0				Reserved for future definition by the ITU-T	0
							x			1 denotes V.34 duplex availability	1
								x		1 denotes V.34 half-duplex availability	2
									1	Stop bit	
0										Octet – 'modn1'	
	x									1 denotes V.32 *bis*/V.32 availability	3
		x								1 denotes V.22 *bis*/V.22 availability	4
			x							1 denotes V.17 availability	5
				0	1	0				Indicates an extension octet	
							x			1 denotes V.29 half-duplex availability (as used in T.30, etc.	6
								x		1 denotes V.27 *ter* availability	7
									1	Stop bit	
0										Octet – 'modn2'	
	x									1 denotes V.26 *ter* availability	8
		x								1 denotes V.26 *bis* availability	9
			x							1 denotes V.23 duplex availability	10
				0	1	0				Indicates an extension octet	
							x			1 denotes V.23 half-duplex availability	11
								x		1 denotes V.21 availability	12
									1	Stop bit	

6.3 Protocols

Table 5 lists codes within the protocol category.

If the LAPM protocol code is indicated in CM and the answer DCE wishes to use LAPM, a protocol octet is also transmitted in JM indicating LAPM.

TABLE 5/V.8

The protocol category

Start	b0	b1	b2	b3	b4	b5	b6	b7	Stop	Octet – 'prot0'
0	0	1	0	1						Tag b0-b3 indicates the protocol category
					0					Indicates a category octet
						1	0	0		Calls for LAPM protocol according to V.42
						1	1	1		Calls for protocol as indicated in an extension octet
									1	Stop bit
NOTE – Absence of this octet does not preclude alternative means of protocol negotiation.										

6.4 GSTN access

Table 6 provides codes for indicating cellular access to the GSTN connection.

TABLE 6/V.8

GSTN access category

Start	b0	b1	b2	b3	b4	b5	b6	b7	Stop	Octet – 'access0'
0	1	0	1	1						Tag b0-b3 indicates the GSTN access category
					0					Indicates a category octet
						x				1 denotes that the call DCE is on a cellular connection
							x			1 denotes that the answer DCE is on a cellular connection
								0		Reserved for future definition by the ITU-T
									1	Stop bit
NOTE – Absence of this octet conveys no information about the type of GSTN access.										

7 Descriptions of signals

7.1 Function indicator signal CI

To initiate a session of data transmission on the GSTN, a DCE transmits either CI, CT, CNG or no signal. Signal CI is a V.8 alternative to call tone CT, and is coded to indicate a call function. The term "call signal" is used hereinafter to refer to CI, CT or CNG.

CI is transmitted from the call DCE with a regular ON/OFF cadence. The ON periods shall be not less than 3 periods of the CI sequence, and not greater than 2.0 s in duration; the OFF periods shall be not less than 0.4 s and not greater than 2.0s in duration.

A CI sequence consists of 10 ONEs followed by 10 synchronization bits and the call function octet.

The transmission and detection of CI is optional in most DCE Recommendations. Whether or not this option is used, DCEs conforming with Recommendation V.8 should not malfunction if CI is received.

7.2 Modified Answer Tone ANSam

Modified answer tone ANSam consists of a sinewave signal at 2100 ± 1Hz with phase reversals at an interval of 450 ± 25 ms, amplitude-modulated by a sinewave at 15 ± 0.1 Hz. The modulated envelope shall range in amplitude between (0.8 ± 0.01) and (1.2 ± 0.01) times its average amplitude. The average transmitted power shall be in accordance with Recommendation V.2.

The average power outside the band 2100 ± 200 Hz produced by using an approximation to the 15 Hz sinewave envelope shall be at least 24 dB below the average power within that band.

When network echo canceller disabling is not required, phase reversals shall not be imparted to the ANSam signal.

A call DCE shall not transmit a signal CM unless ANSam has been detected.

NOTE – The call DCE needs to distinguish ANSam from ANS. Detector design needs to allow for transient variations in the received answer-tone amplitude and phase that may be generated occasionally by network equipment.

7.3 The Call Menu signal CM

Signal CM initiates the process of modulation-mode selection.

A CM sequence starts with 10 ONEs followed by 10 synchronization bits as given in Table 1.

The first information category in CM indicates the required call function in accordance with Table 3. CM shall also include one or more octets indicating available modulation modes in accordance with Table 4.

The protocol category may be included in order to negotiate LAPM without requiring the ODP/ADP exchange (see 7.2.1/V.42) (see Table 5).

The GSTN access category is included if the call DCE wishes to indicate cellular access (see Table 6). In this case bit b5 is set to ONE and bit b6 is set to ZERO.

A CM signal is terminated (after JM detection) by the transmission of a CJ signal.

7.4 The Joint Menu signal JM

A signal JM is transmitted from an answer DCE only in response to a detected CM signal. JM shall be transmitted after a minimum of two identical CM sequences have been received.

A JM sequence starts with 10 ONEs followed by 10 synchronization bits as given in Table 1.

The first information category in JM indicates the same call function as in the received CM or, if the call function is not available in the answer DCE, JM may indicate a different call function (see 8.2.3).

If there are modulation modes in common between the call and answer DCEs, JM shall include the octets necessary to indicate all modulation modes that are both indicated in CM and available in the answer DCE for use with the call function indicated in CM. Additional modulation mode octets that are in CM may also be included. The indicated modulation mode with the lowest item number (see Table 4) shall be used in the subsequent data session.

If there are no modulation modes in common between the call and answer DCEs, the JM sequence shall include the same number of modulation-mode octets as CM, and show zeros for all modulation modes.

If the LAPM protocol code is indicated in CM, the protocol octet may be included in JM in order to complete the negotiation of LAPM (see Table 5).

The GSTN access category is included if the answer DCE wishes to indicate cellular access, or if this category is present in the received CM (see Table 6). If the answer DCE wishes to indicate cellular access, bit b6 is set to ONE. Bit b5 is set to ONE if and only if the corresponding bit (b5) is set to ONE in the received CM.

8 Data session start-up procedure

Figure 4 shows the signal interaction diagram with CI, ANSam and CM/JM signals. T_e is the silent period allowed for disabling of network echo-control equipment.

8.1 Start-up procedure in the call DCE

8.1.1 Call signal transmission

After transmitting no signal for 1 s, the DCE shall initiate transmission of CI, CT or CNG, or continue transmission of no signal.

The DCE shall then seek to detect ANS, ANSam, or a sigA that is characteristic of an acceptable mode of modulation.

NOTE – The transmission of a V.21(H) signal before the transmission of answer tone is being studied by the ITU-T for some facsimile applications. A call DCE conforming with Recommendation V.8 should not malfunction if such a signal is received.

If a suitable sigA is detected, then the call modem shall proceed in accordance with the modulation mode indicated by sigA. Such procedure is outside the scope of this Recommendation.

After detection of ANS or ANSam, the call signal shall be stopped. However the call DCE may choose to ensure that CI has been transmitted for a minimum of 3 full sequences.

If ANSam (rather than ANS) is detected, the DCE shall transmit no signal for a period T_e prior to transmitting signal CM. The silent period T_e begins after the termination of the call signal or, in the absence of a call signal, after the detection of ANSam. The minimum value for T_e shall be 0.5 s. However, if it is desired to allow for network echo canceller disabling in the manner defined in Recommendation V.25, T_e shall be set to a value ^{3} 1s. The procedure shall continue in accordance with 8.1.2.

If ANS (rather than ANSam) is detected, the DCE shall proceed in accordance with Annex A/V.32 *bis*, Recommendation T.30, or other appropriate Recommendation.

8.1.2 Call menu transmission

When interval T_e has elapsed, the call DCE shall initiate transmission of signal CM and condition its receiver to detect signal JM.

After a minimum of 2 identical JM sequences have been received, the call DCE shall complete the current octet and associated start and stop bits and then signal CJ shall be transmitted. Following CJ, the call DCE shall transmit no signal for a period of 75 ± 5 ms, transmit sigC and proceed in accordance with the selected V-Series modulation mode.

If JM showed zeros for all modulation modes, the call DCE may disconnect after transmission of CJ.

8.2 Start-up procedure in the answer DCE

For a period of at least 0.2 s after connection to line, the answer DCE shall transmit no signal.

Some Recommendations require that the answer DCE shall delay the transmission of answer tone unless or until some signal is detected from the call DCE. In these cases, upon detection of CI, CT or CNG as appropriate, the DCE shall proceed in accordance with 8.2.1 or 8.2.2.

Other Recommendations require that answer tone shall be transmitted without waiting for call signals. Such DCEs shall proceed directly in accordance with 8.2.1 or 8.2.2.

8.2.1 ANS transmission

Some Recommendations require the transmission of unmodulated answer tone (ANS) and do not allow for CM/JM exchanges. The procedure following the transmission of ANS is outside the scope of this Recommendation.

8.2.2 ANSam transmission

If the answer DCE supports CM/JM exchanges, ANSam shall be transmitted.

Upon receiving a minimum of 2 identical CM sequences, the DCE shall transmit JM and proceed in accordance with 8.2.3.

If a suitable sigC is detected during ANSam transmission, the DCE shall transmit no signal for 75 ± 5 ms, transmit the appropriate sigA and continue in accordance with the relevant modem Recommendation.

If neither CM nor a suitable sigC is detected during ANSam transmission, the DCE shall transmit no signal for 75 ± 5 ms and then continue in accordance with Annex A/V.32 *bis*, or Recommendation T.30 or other appropriate Recommendation. If not terminated by the receipt of CM or a suitable sigC, ANSam shall be transmitted for a period of 5 ± 1 s.

8.2.3 JM transmission

If the call function is available, JM shall be coded to indicate the same call function as CM, and to indicate the modulation modes jointly available in call and answer DCEs.

If the call function is not available, the answer DCE may indicate an available call function different from CM. If JM is sent, it shall include the same number of modulation mode octets as CM and show zeros for all modulation modes.

JM transmission shall continue until signal CJ is detected and all 3 octets of CJ have been received. In the case that CJ is not correctly received, other criteria may be used to terminate transmission of JM, such as detection of sigC corresponding to the selected modulation mode, or the absence of CM for a suitably long period of time.

JM shall be terminated without any requirement to complete a current JM sequence. No signal shall be transmitted for a period 75 ± 5 ms, followed by sigA corresponding to the selected modulation mode. SigA and subsequent responses shall be as defined in the relevant V-Series modem Recommendation.

If JM shows zeros for all modulation modes, the answer DCE may disconnect on reception of CJ.

9 DTE-DCE interchange circuits

During the V.8 procedure, there is no requirement for DTE-DCE communication, and the Recommendation does not provide for any such communication. The states of interchange circuits may therefore be determined by the procedures before and after the V.8 procedure.

According to the present invention, the use of an etiquette for identifying the type of call can be used in various telephone systems. These include the following:

A. CMA based system

In such a system in the (first stage one call tooriginating ISP 106A algorithms to pick based on cost of QoS); in the second stage a CMA System disclosed in co-assigned US Patent Applications Nos. 08/731,848 and entitled A WAN Based System and Method for Personal Communication, filed October 21, 1996, and in continuation-in-part of US Patent Application No. 08/731,848 filed on January 20, 1997, identifies answering end ISP router (same algorithms); in the third stage PSTN calls to answering modem (means to bill back call to originating party ISP account possibly using Called Line ID). At the answering end V.8 or V.8bis negotiated audio answering could also support all possible types of audio messaging the PC provides if the user was not present.

B. Operator assisted Internet access

Once voice over the Internet use was wide spread and assuming that Internet calls were much lower cost than circuit mode calls then a simple box that attached to any telephone line electrical interface-line powered or audio interface ([rubber cup]-battery powered) would be useful for non-computer user wishing easily to make low cost telephone calls. Such a box could store the addressing needed and the addressing could be designated via the telephone key pad or a push button on the box and sent with the appropriate etiquette signaling to the nearest desired ISP.

A simple acoustic coupled box is transmit only. The originating user upon hearing answer tone from the ISP sends V.8CI with the unique call function followed by needed information fields coded in a proprietary fashion. Alternatively, the originating user upon hearing answer tone from the ISP sends ES MS of V.8bis with the Non Standard Field or Network function followed by needed information fields coded in a proprietary fashion.

According to yet another aspect of the invention the access time to the INTERNET, termed INTERNET Access Time (IAT) is reduced as described below to enable faster connection for any of the communications described above as well as for any other INTERNET connection.

Presently, total IAT time is comprised from the following steps done in sequence and described in Figure 3A:

1. Telephone call progress. This stage is the time it takes from end of dialing by the initiating modem to the first response by the responding modem at the receiving end.
2. V.8 or V.8 bis and network interaction. This stage is the initial interaction between the initiating modem and the receiving modem. This stage is carried out in accordance with Phase 1 of the V.34 protocol.
3. INFO exchanges probing/ranging. This stage is an information exchange between the modems, for example rate matching. This stage is carried out in accordance with Phase 2 of the V.34 protocol.
4. Equalizer and Echo Canceling Training. This stage which usually is the longest stage of the modem start-up (see below) takes few seconds (typically 5 - 8 seconds). In this stage echoes over the communication line are minimized. This stage is carried out in accordance with Phase 3 of the V.34 protocol.
5. Final training. This stage is the finalize the modem start-up stage and is carried out in accordance with Phase 4 of the V.34 protocol.

   Steps 2 - 5 are also known collectively as modem start-up and are shown as one step in Figure 3A.
6. ASCII log-in (nominal 25 bytes). In this stage connection to the INTERNET starts with user name log-in.
7. PPP initialization. In this stage the low level Point to Point Protocol (PPP) is initiated to start low level connection (92 bytes + options) with the INTERNET. Usually the user connects at this stage to is INTERNET Access Provider server.

8. PAP (Password Authentication Protocol) or CHAP (Challenge Handshake Authentication Protocol). At this stage authentication of the user password takes place and the PPP low level protocol is replaced by a medium level protocol (PAP or CHAP) operating in the several hundred bytes range.
9. Access higher layer resources (TCP/IP set-up, proxy server, RSVP). In this stage the high level protocol used during actual transmission of data over the INTERENET (usually TCP/IP) is activated and connection to the INTERNET is completed.

Steps 6 - 9 will be termed herein collectively as actual access stage.

According to the present invention, some of the steps described below, presently done in sequence are done in parallel, thus reducing the IAT. According to a preferred embodiment illustrated schematically in Fig. 3B, step 6 (ASCII log in) is done during step 2 of the modem start-up stage (V.8 or V.8 bis and network interaction). Sending the ASCII log-in sequence during stage 2 offers the prospect of reducing the IAT by the total modem start-up time. This occurs because the two longest/most common temporal components of the IAT are the modem start-up (all 4 phases) and the ASCII log-in authentication. Paralleling modem start-up and ASCII authentication offers significant IAT savings.

The implementation of this change via changes in V.8 are provided in Appendix B.

APPENDIX B

Current use of the NS field in V.8 (SG16 March 1997 TD-92 PLEN)

The following provides the details to implement non-standard information fields in V.8 CM/JM exchanges.

Add the V.8 (Table 2) category octet 11110xxx (b0 to b7, left to right) Non-Standard (NS) Field.

Add to V.8 par 6.5 Non-Standard Information Block, as follows

CM and JM messages may optionally include a non-standard information filed following the standard fields in each CM/JM sequence to define information beyond that defined in this Recommendation. When non-standard information is to be sent, the NS Field category octet is set to 11110xxx.

Each non-standard information block is composed of:

Non-Standard Field length K+L+M+1 (1octet)
T.35 Country code (K octets)
Provider Code length=L (1 octect)
T.35 Provider Code (L octets)
Non-Standard Information

The NS field is parsed in accordance with the rules for extension octets in paragraph 5.2. This distributes each five bits of NS (higher order NS bits in higher order b positions) over 10 bits of extension octets.

Multiple concatenated NS information blocks may be transmitted.

Discussion

The above V.8 NS mechanisms can be enhanced in a number of ways:

1. Support NS with optional CI as well as CM/JM to initiate ISP "pre-announce" procedures as early in the V.8 negotiation as possible.
2. Reduce the coding constrain for NS with optional CI from 5 useable bits of every 10 to 8 usable bits of every 10. This supports additional

information to allow a range of non-standard ISP "pre-announce" procedures.

3. Add a CRC error checking sequence to CI/NS messages to avoid the requirement to receive 300 bit/s NS messages multiple times.
4. Use an accept/reject JM NS information category response to avoid a repeat transmission of the received NS at 300 bit/s

Proposed Wording

This proposed wording provides the details of the changes required to V.8 to implement the four enhancements discussed above. It assumes the changes described in TD-92 PLEN have been included. It does not encompass a redraft of V.8 to make all related wording changes that would be desirable to the Recommendation.

7.1 Function indicator signal CI

<replace the second and third paragraphs with the following:>

A CI sequence consists of 10 ONEs followed by 10 synchronization bits and a call function or an information category octet followed by other octets as defined below. CI is transmitted from the call DCE with a regular ON/OFF cadence in one of two modes:

1. When CI is used with a call function octet, no other octets may follow, the ON periods shall not be less than 3 periods of the CI sequence, and not greater than 2.0s in duration: the OFF periods shall not be less than 0.4s and not greater than 2.0s in duration.
2. When CI is used with an information category octet, other octets may follow, the ON periods are not greater than 2.0s in duration: the OFF periods shall not be less than 0.4s and not greater than 2.0s in duration.

A call function octet and an information category octet cannot be transmitted in the same repeating CI sequence.

<Add to V.8 par 6.5 Non-Standard Information Field, as follows:>

Table 7/V.8 provides NS codes for the transmission of JM responses to CI with NS signals.

Start	b0	b1	b2	b3	b4	b5	b6	b7	stop	Octet Non-Standard (NS)
0	1	1	1	1						Tag b0-b3 indicates NS octet
					0					Indicates tagged category octet
						1	0	1		Indicates acceptance of NS
						0	1	0		Indicates rejection of NS
						1	1	1		Indicates conditional acceptance of NS
									1	Stop bit

Conditional acceptance of NS indicates that Country code and Provider code are accepted, but that some aspect of the Non-Standard Information is not accepted. Procedures to follow in this case are beyond the scope of this Recommendation.

<add the following last paragraph to 7.4 The Joint Menu signal JM:>

Table 7/V.8 provides NS codes for the transmission of JM responses to CI with NS signals. When a CI with information category is received the JM response, after the CM is received, will include the appropriate code from Table 7.

<Add to V.8 par 6.5 Non-Standard Information Block, as follows:>

CI messages may optionally include a non-standard (NS) information field to define information beyond that defined in this Recommendation. When non-standard information is to be sent in the CI, the NS Field category octet is set to 11110xxx.

Each CI, NS information clock is composed of:

Non-Standard Field length K+L+M+3 (1octet)
T.35 Country code (K octets)
Provider Code length=L (1 octect)
T.35 Provider Code (L octets)
Non-Standard Information
Frame check sum (tbd)

The CI/NS field is parsed in accordance with the rules for extension octets in paragraph 5.0. This allows eight bits of information bearing NS (higher order NS bits in higher order b positions) over each 10 bits as CI uses V.21 (L) modulation to avoid confusion with T.30 signals.

The maximum length of the non-standard information block when used with CI is 57 octets to prevent the CI + information block from exceeding 2 seconds.

It will be appreciated that IAT saving is not limited to INTERNET Access Time and may be extended to INTRANET Access Time. According to one non limiting example illustrated in Fig. 3C, PAP or CHAP authentication, which occurs over a PPP link, is optional and usually utilized for INTRANET access. Sending PAP or CHAP authentication using extensions to the INFO sequences (600 bit/s) or possibly a higher rate data exchange during phase 2 would significantly speed the access to INTRANETS, such as corporate INTRANETS. In effect this would parallel PAP or CHAP authentication with the ASCII authentication. Still further, instigating higher layer resources also has the potential to save significant IAT. In effect this would parallel PAP or CHAP negotiation with the negotiation of higher layer resources

It will be appreciated by persons skilled in the art that the present invention is not limited by what has been particularly shown and described herein above. Rather the scope of the invention is defined by the claims which follow:

CLAIMS

1. A method for audio communication over an integrated STN-computer network comprising:

   identifying an originated audio communication as either an analog audio communication or an audio over data path communication; and

   transmitting said audio communication in accordance with its identification.

2. A method according to claim 1 wherein said identifying comprising employing a communication protocol utilized to establish said audio communication to provide said identification.

3. A method according to claim 2 wherein said protocol is the V.8 or V.8bis protocol.

4. A system for audio communication over an integrated STN-computer network comprising:

   a. a plurality of audio communication devices;

   b. local and remote Internet Service Providers (ISP) connected to each of said plurality of audio communication devices by an STN portion;

   c. a data packet based Internet portion connecting said local and remote Internet Service Providers (ISP), and

   d. identification means for identifying said plurality of audio communication devices, as either an analog audio communication or an audio over data path communication, to said local and remote ISP.

5. A system according to claim 4 wherein said identification means comprises a communication protocol.

6. A system according to claim 5 wherein said communication protocol comprises a V.8 or V.8bis protocol.

7. A system according to claim 6 wherein the signals of said V.8 or V.8bis are generated by said audio communication device or by a dedicated device.

8. A method for saving INTERNET Access Time comprising the step of performing at least part of the modem start up stages with at least part of the actual access stage.

9. A method according to claim 8 wherein the ASCII log stage of said actual access steps is performed at least partially concurrently with the network interaction step of said modem start-up stage.

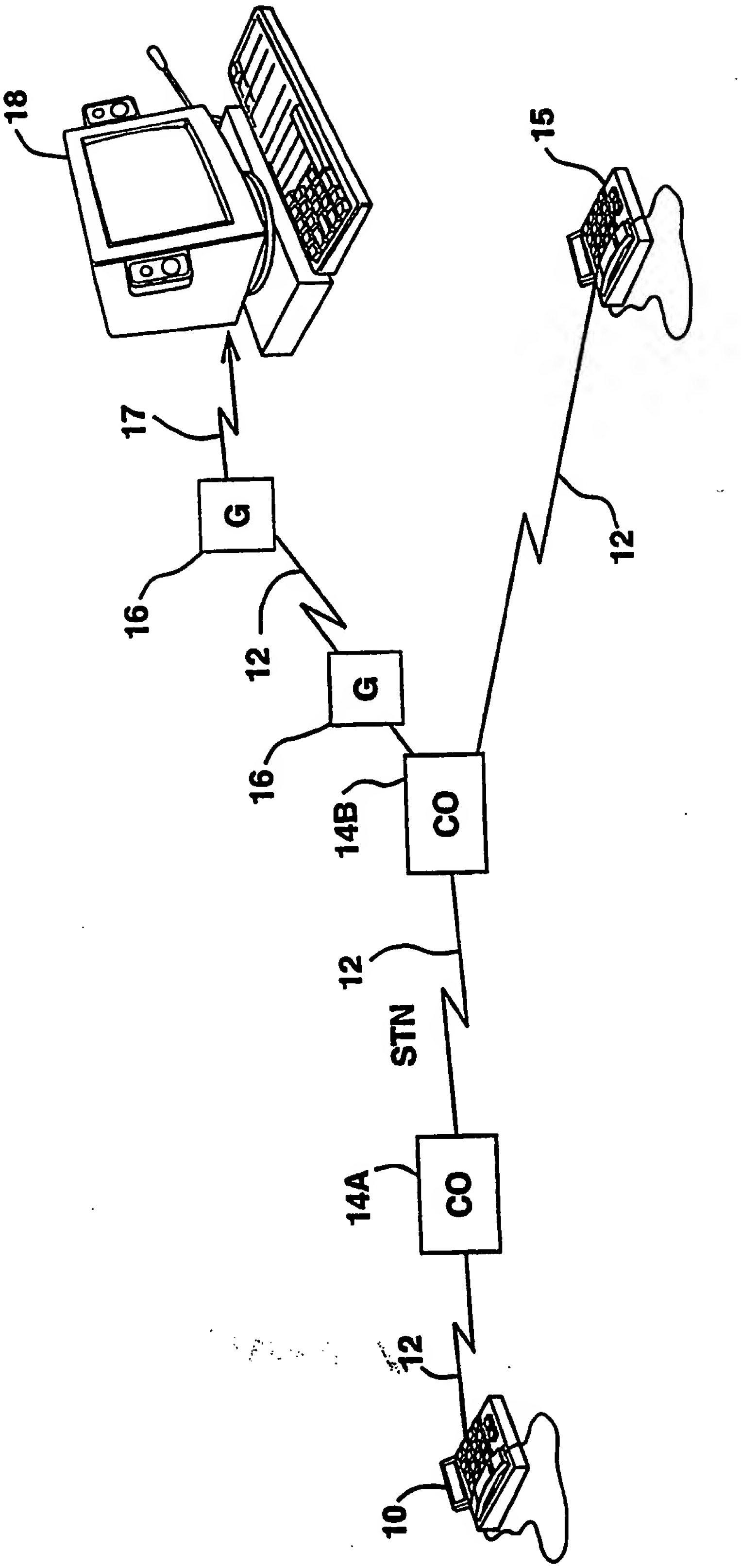


FIG. 1 (Prior Art)

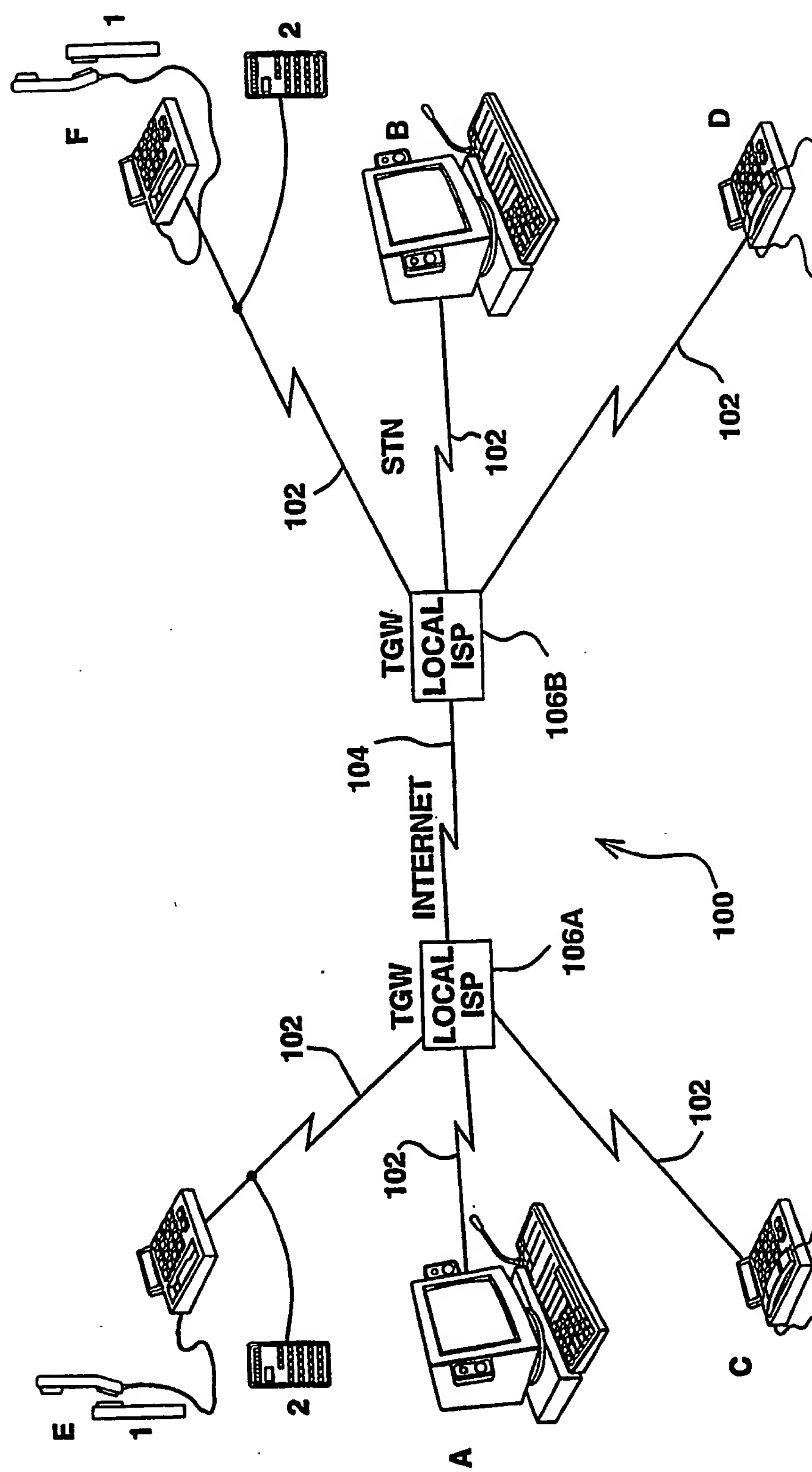

1
2
F
B
D
STN
102
102
102
102
TGW
LOCAL
ISP
106B
104
INTERNET
100
TGW
LOCAL
ISP
106A
102
102
102
E
1
2
A
C

FIG. 2

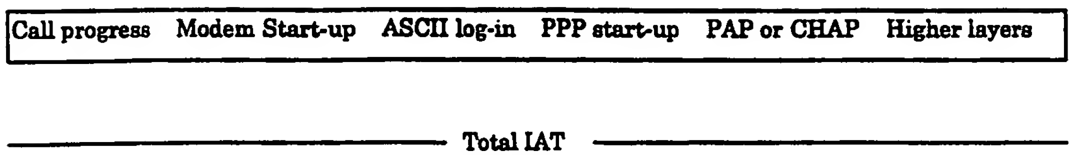


FIGURE 3A (PRIOR ART)

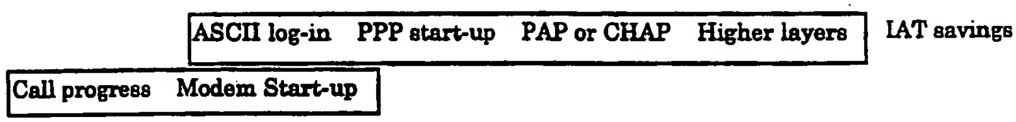


FIGURE 3B

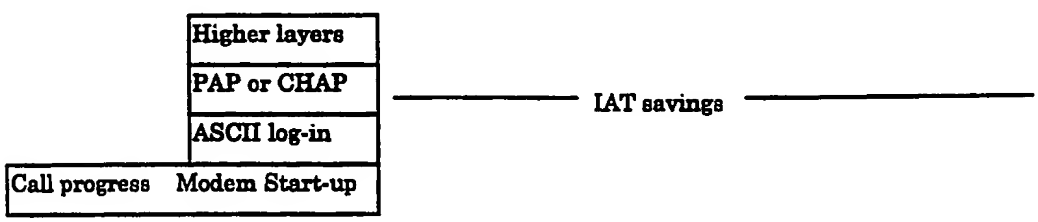
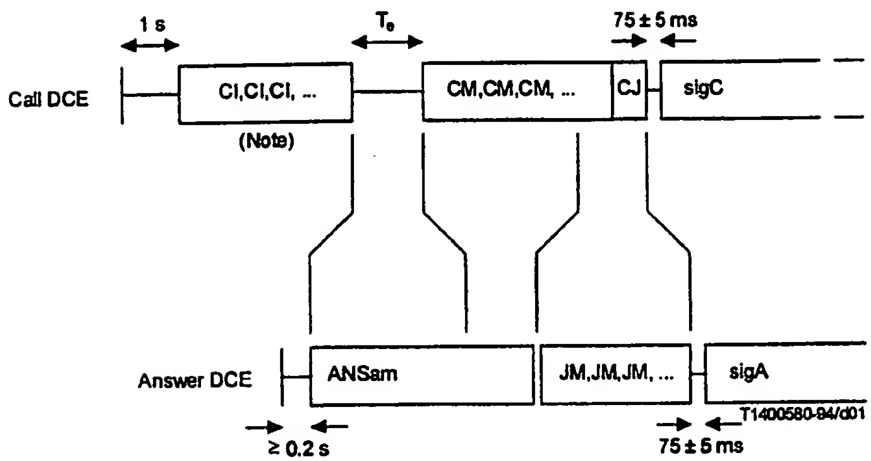


FIGURE 3C



NOTE – Use of CI as a call signal is optional. Compatibility with existing answer terminals will sometimes mandate the use of CNG or CT.

FIGURE 4

Use of the CI call signal and exchange of CM/JM menu signals

INTERNATIONAL SEARCH REPORT

International application No.
PCT/IL98/00122

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :H04L 12/28
US CL : 370/389
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 370/389, 364, 365

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

APS

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X,E --- Y,E	US 5,761,294 A (SHAFFER et al) 02 June 1998, Figs. 1, 3, 4; col. 3, lines 27-60; col. 4, lines 40-55.	8-9 --- 1-7
Y,P	US 5,721,731 A (YOSHIDA) 24 February 1998, abstract; fig. 1; col. 2, lines 52-65.	1-7

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search

05 JUNE 1998

Date of mailing of the international search report

12 AUG 1998

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